We Claim:

1. A software radio port device for processing Voice over IP (VoIP) data packets for wireless terminals, the device comprising:

an air interface;

an IP/Ethernet Interface;

a VoIP Media Gateway interposed between the air interface and the IP/Ethernet Interface for media conversion and transportation;

a VoIP signaling Gateway for controlling VoIP call processing; and,

a Call Control for controlling call processing of wireless terminals and coordinating with VoIP call processing.

2. The device of claim 1 wherein the air interface receives signaling messages and a voice stream from a mobile station.

2.5. The device of claim 1 wherein the mobile station receives signaling messages and a voice stream from the air interface.

3. The device of claim 1 wherein the Call Control is configured for:

receiving signaling messages from the air interface;

instructing the VoIP Media Gateway to set up RTP paths to the called parties; and,

instructing the VoIP Signaling Gateway to set up VoIP calls to the called parties.

4. The device of claim 1 wherein the VoIP Signaling Gateway is configured for:

receiving messages from the Call Control;

processing messages from the Call Control; and,

managing VoIP call-related activities.

5. The device of claim 1 wherein the VoIP Media Gateway is configured for:

receiving messages from the Call Control;

processing messages from the Call Control;

receiving the voice stream from the air interface; and
packetizing the voice stream into RTP data packets.

6. The device of claim 1 wherein the IP/Ethernet Interface receives RTP data packets from the VoIP Media Gateway and messages from the Call Control and VoIP Signaling Gateway, and sends the RTP data packets and the messages to the packet data network.

7. The device of claim 1 wherein the IP/Ethernet Interface receives messages and RTP packets from a packet data network, sends the RTP packets to the VoIP Media Gateway, and sends the messages to the Call Control and VoIP Signaling Gateway.

8. The device of claim 1 wherein the Call Control is configured for:
receiving signaling messages from the IP/Ethernet interface and the VoIP Signaling Gateway; and
managing mobile station-related activities.

9. The device of claim 1 wherein the VoIP Signaling Gateway is configured for:
receiving messages from the IP/Ethernet interface;
instructing the Call Control to manage mobile calls; and
managing VoIP call-related activities.

10. The device of claim 1 wherein the VoIP Media Gateway is configured for:
receiving the VoIP data packets from the IP/Ethernet Interface; and
converting the VoIP data packets to a voice stream.

11. The device of claim 1 wherein the Air Interface receives a voice stream from the VoIP Media Gateway and receives signaling messages from the Call Control.

12. A wireless telecommunication system for providing VoIP service to wireless terminals comprising:

a software radio port device according to claim 1;

a network server platform;

a VoIP-enabled communication device;

a VoIP proxy server for managing requests/messages from the VoIP-enabled communication device; and

a PSTN/VoIP Gateway for interconnecting a VoIP network with Public Switched Telephone Network (PSTN).

13. The system of claim 12 wherein the network server platform comprises:

an IP interface;

a Mobile Switching Center/Visitor's Location Register (MSC/VLR) configured to provide call control operations for mobiles;

a Home Location Register (HLR) for storing mobile subscriber authentication data; and

a VoIP Call-Server Control to provide VoIP call processing control operations.

14. A method of providing a two-way voice path between a VoIP device in a network and a mobile station wherein a call originates at the mobile station, the method comprising:

initiating mobile call set-up;

tuning the mobile station to digital traffic channel (DTC) to establish a voice path over the air;

engaging a VoIP call-server to set up a VoIP call;

generating a ringback tone to the mobile station;

establishing an RTP media path for exchange of RTP data packets; and

interconnecting the voice path over the air and the RTP path over the packet network.

14.5. The method of claim 14 wherein the VoIP device comprises at least one of a VoIP phone or a VoIP Gateway.

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15. The method of claim 14 wherein the step of initiating mobile call set-up comprises receiving a call origination message from the mobile station and engaging the NSP to set up a mobile call.

16. The method of claim 14 wherein said step of tuning comprises:
sending a message to tune the mobile station to a specified digital traffic channel; and
detecting the mobile station as being tuned to the specified digital traffic channel.

17. The method of claim 14 wherein said step of engaging comprises:
sending a VoIP call connection request to the VoIP call-server;
analyzing a called number; and
setting up a VoIP call via the VoIP call-server.

18. The method of claim 14 wherein said step of generating comprises:
receiving a ringing indication from the called party;
generating a ringback tone in response to said receiving; and
transmitting the ringback tone to the mobile station.

19. The method of claim 14 wherein said step of establishing comprises:
receiving a connect indication from the called party;
turning off the ringback tone;
setting up an RTP media path for exchange of RTP data packets; and
informing the NSP of the call connection.

20. The method of claim 14 wherein said step of interconnecting comprises:
converting received voice frames to RTP packets to be sent to the packet network, and
converting received RTP packets to voice frames to be sent to the mobile station.

21. The method of claim 14 wherein the network is a Public Switched Telephone Network (PSTN).

22. The method of claim 14 wherein the network is a Private Branch Exchange (PBX).

23. A method of providing a two-way voice path between a VoIP device in a network and a mobile station wherein a call originates at the VoIP device, the method comprising:

processing a call connection request at a VoIP call-server;

initiating mobile call set-up at a Network Server Platform (NSP);

tuning the mobile station to a digital traffic channel (DTC) to establish a voice path over the air via a Software Radio Port (SRP);

alerting both the mobile station and the VoIP device;

establishing an RTP media path for exchange of RTP data packets via the SRP; and

interconnecting the voice path over the air and the RTP path over the packet network via the SRP.

24. The method of claim 23 wherein the step of processing a call connection request comprises receiving a call connection request message from the VoIP device and engaging an NSP to analyze the called number.

25. The method of claim 23 wherein said step of initiating mobile call set-up comprises:

verifying the called party as a valid mobile station;

sending a message to page the mobile station via the SRP;

receiving page response from the mobile station; and

instructing the VoIP call-server to forward the call connection request to the SRP.

26. The method of claim 23 wherein said step of tuning comprises:

sending a message to tune the mobile station to a specified digital traffic channel; and

detecting the mobile station as being tuned to the specified digital traffic channel.

27. The method of claim 23 wherein said step of alerting comprises:

sending a message to the mobile station for alerting a mobile user; and

sending a ringing indication to the VoIP device via the VoIP call-server.

28. The method of claim 23 wherein said step of establishing comprises:
receiving a connect indication from the mobile station;
sending the connect indication to the VoIP device via the VoIP call-server;
setting up an RTP media path for exchange of RTP data packets; and
informing the call connection to the NSP.

29. The method of claim 23 wherein said step of interconnecting comprises:
converting received voice frames to RTP packets to be sent to the packet network, and
converting received RTP packets to voice frames to be sent to the mobile station.

30. The method of claim 23 wherein the network is a Public Switched Telephone Network (PSTN).

31. The method of claim 23 wherein the network is a Private Branch Exchange (PBX).

32. A method of providing a two-way voice path between a first mobile station and a second mobile station wherein the first mobile station is associated with a first Software Radio Port (SRP) and the second mobile station is associated with a second SRP and wherein a call originates at the first mobile station, the method comprising:

initiating call set-up for the first mobile station at the first SRP;

tuning the first mobile station to a digital traffic channel (DTC) via the first SRP to establish a voice path over the air;

engaging a VoIP call-server to set up a VoIP call via the first SRP;

initiating mobile call set-up for the second mobile station via a Network Server Platform (NSP);

tuning the second mobile station to a digital traffic channel (DTC) via the second SRP to establish a voice path over the air;

alerting the first mobile station and the second mobile station via the second SRP;

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generating a ringback tone to the first mobile station via the first SRP;

establishing an RTP media path for exchange of RTP data packets;

interconnecting a voice path between the first SRP and the first mobile station and an RTP path over the packet network; and

interconnecting a voice path between the second SRP and second mobile station and an RTP path over the packet network.

33. The method of claim 32 wherein the step of initiating call set-up for the first mobile station at the first SRP comprises receiving a call origination message from the first mobile station and engaging the NSP to set up a mobile call.

34. The method of claim 32 wherein said step of tuning the first mobile station to a digital traffic channel (DTC) via the first SRP comprises:

sending a message to tune the first mobile station to a specified digital traffic channel; and

detecting the first mobile station as being tuned to the digital traffic channel.

35. The method of claim 32 wherein said step of engaging a VoIP call-server to set up a VoIP call comprises:

sending a VoIP call connection request to the VoIP call-server; and

coordinating with the NSP to analyze a called number via the VoIP call server.

36. The method of claim 32 wherein said step of initiating mobile call set-up for the second mobile station via a Network Server Platform comprises:

verifying the called number as a valid mobile station;

sending a message to page the second mobile station via the second SRP;

receiving a page response from the second mobile station; and

instructing the VoIP call-server to forward the call connection request to the second SRP.

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37. The method of claim 32 wherein said step of tuning the second mobile station to a digital traffic channel (DTC) via the second SRP comprises:

sending a message to tune the second mobile station to a specified digital traffic channel; and

detecting the second mobile station as being tuned to the specified digital traffic channel.

38. The method of claim 32 wherein said step of alerting the first mobile station and the second mobile station comprises:

sending a message to the second mobile station for alerting a user; and

sending a ringing indication to the first SRP via the VoIP call-server.

39. The method of claim 32 wherein said step of generating a ringback tone to the first mobile station via the first SRP comprises:

receiving a ringing indication from the called party;

generating a ringback tone in response to said receiving; and

transmitting the ringback tone to the first mobile station.

40. The method of claim 32 wherein said step of establishing an RTP media path for exchange of RTP data packets comprises:

receiving a connect indication at the second SRP from the second mobile station;

sending a connect indication from the second SRP to the VoIP call-server;

receiving a connect indication at the first SRP from the VoIP call-server;

sending back an acknowledge message from the first SRP;

turning off the ringback tone;

setting up the RTP media path for exchange of RTP data packets; and

informing the NSP of the call connection.

41. The method of claim 32 wherein said step of interconnecting a voice path between the first SRP and the first mobile station and an RTP path over the packet network comprises:

converting received voice frames from the first mobile station to RTP packets to be sent to the packet network; and

converting received RTP packets to voice frames to be sent to the first mobile station.

42. The method of claim 32 wherein said step of interconnecting a voice path between the second SRP and second mobile station and an RTP path over the packet network comprises:

converting received voice frames from the second mobile station to RTP packets to be sent to the packet network; and

converting received RTP packets to voice frames to be sent to the second mobile station.

43. A method for terminating a call between a mobile station and a VoIP device in a network comprising:

receiving a release indication from the mobile station;

releasing radio resources and an RTP media path;

sending a call release request to the VoIP device via a VoIP call-server; and

sending a call release indication to a Network Server Platform (NSP).

44. A method for terminating a call between a mobile station and a VoIP device in a network comprising:

receiving a release indication from the VoIP device via a VoIP call-server;

sending a call release request to the mobile station;

releasing radio resources and an RTP media path; and

sending a call release indication to a Network Server Platform (NSP).

45. A method for terminating a call between a first mobile station and a second mobile station, said first mobile station associated with a first Software Radio Port (SRP) and said second mobile station associated with a second SRP, the method comprising:

receiving a release indication at the first SRP from the first mobile station;

releasing radio resources and an RTP media path at the first SRP;

sending a call release request from the first SRP to a VoIP call-server;

sending a call release indication from the first SRP to a Network Server Platform (NSP);

receiving a release indication at the second SRP from the VoIP call-server;

sending a call release request from the second SRP to the second mobile station;

releasing radio resources and an RTP media path at the second SRP; and

sending a call release indication from the second SRP to NSP.

46. A method for maintaining an RTP media path during handoff of a mobile station from a first Software Radio Port (SRP) to a second Software Radio Port (SRP) wherein the mobile station is connected with a party, the method comprising:

sending a handoff request from the first SRP to a Network Server Platform (NSP);

handing off the mobile station from the first SRP to the second SRP via the NSP;

sending a call transfer request from the first SRP to the NSP;

releasing radio resources at the first SRP;

detecting at the second SRP the mobile station as being tuned to a digital traffic channel and sending a conference call request to the party via a VoIP call-server;

setting up an RTP media path for exchange of RTP data packets via the second SRP when the conference call has been established;

interconnecting the voice path between the second SRP and the mobile station and the RTP path;

sending a handoff complete indication from the second SRP to the NSP;

sending a call release request from the first SRP to the party via the VoIP call-server;

releasing the RTP media path at the first SRP; and

sending call release indication from the first SRP to the NSP.